



Avaya Solution & Interoperability Test Lab

Application Note for ASC Marathon Evolution Call Recording Solution with Avaya Integral 55 – Issue 1.0

Abstract

These Application Notes describe the configuration steps for the ASC Marathon Evolution and the Avaya PBX Integral 55 to interoperate successfully.

ASC Marathon Evolution is a call recording solution for capturing telephone calls from the Integral 55 recording of hardware channels and IP-sniffing of RTP packets. ASC Marathon Evolution uses Computer Telephony Integration (CTI) to extract and process call event information.

Information in these Application Notes has been obtained through compliance testing and additional technical discussions. Testing was conducted via the Developer*Connection* Program at the Avaya Solution and Interoperability Test Lab

1. Introduction

These Application Notes describe the compliance-tested configuration using an ASC Marathon Evolution server, an ASC CTI Controller server, an Avaya CTI server and two Avaya Integral 55 (I55) PBXs.

An Avaya PBX Integral 55 with software version L021 running on an Advanced Computer Board (ACB) was used as hosting PBX for the ASC Marathon Evolution system. Another I55 with software version L021 running on an ACB was linked via an IP-QSIG trunk to test networking scenarios.

Figure 1 shows the integration of the Marathon Evolution system into a network of two Integral 55 PBXs.

The ASC Marathon Evolution consists of two major components: the ASC Marathon Evolution server and the ASC CTI Controller server. ASC CTI Controller server holds the ASC CTI software; ASC Marathon Evolution server holds the voice recording software.

For extension side recording the ASC Marathon Evolution server supports two methods:

- Recording of TDM hardware channels and
- IP-sniffing of RTP packets on the LAN.

The recording of hardware channels is achieved by copying the extensions' voice data (B-channel) onto a distinct timeslot of the PCM30 trunk (consists of 30 B-channels) that is supplied by the MAC board of the Integral 55. The ASC Marathon Evolution server gets these data and is able to record all telephone calls for the corresponding extension.

For the IP-sniffing method the IP-phones have been connected to a Hub as well as the ASC Marathon Evolution server. All RTP packets that contain voice data for the IP-phones are delivered to ASC Marathon Evolution server for recording too.

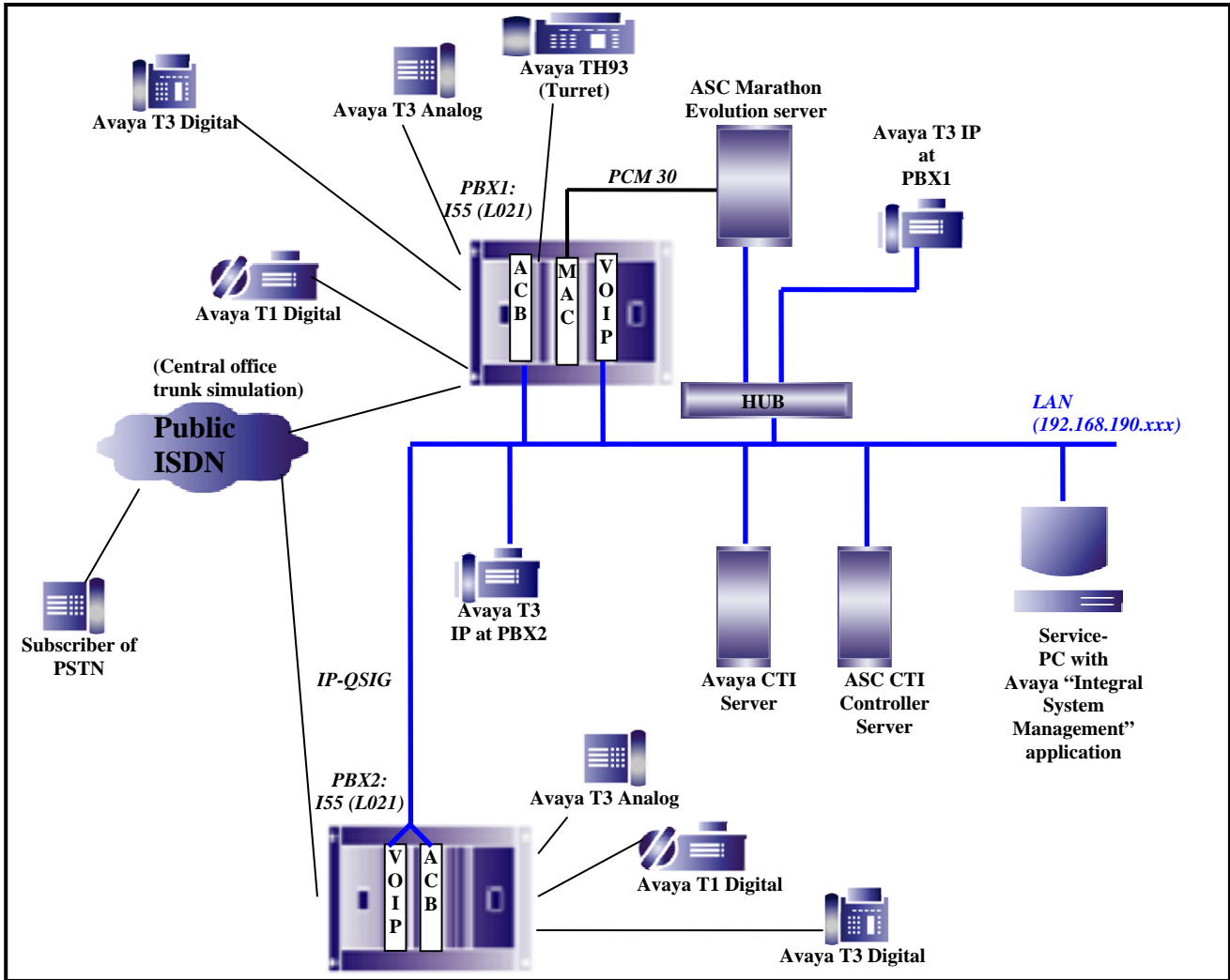


Figure 1: Tested Avaya Integral 55 PBX with ASC Marathon Evolution System

2. Equipment and Software Validated

The following equipment and software were used for the sample configuration provided:

| Equipment | Software/Firmware |
|--|--|
| Integral 55, W1-Module, with MAC circuit pack, providing an E1-trunk interface (PCM30 signal, consisting of 30-B-channels) and with VoIP circuit pack for IP extensions and IP trunks and with DT0 circuit pack for establishing a S0-loop | L021V00_1_0_5.2 |
| Integral 55, W2-Module with VoIP circuit pack for IP extensions and IP trunks | L021V00_1_0_5.2 |
| Avaya CTI Server | Windows 2003-Server, Standard Edition Service Pack 1, with ConneCTIon (CTI-Server and CTI-Administrator), version V3.1.033 |
| ASC CTI Controller Server | 3.0.09 |
| ASC Marathon Evolution server | 6.10.11 |
| Avaya T3 Digital telephones (T3.11_comfort_upn) | Bootloader: V00.09 Software: T314_0DE.hx1 |
| Avaya T3 IP telephones (T3_IP2_comfort) | Bootloader: B01.03 Software: T323_0DE.a3i |
| Avaya T3 Analog telephones (T3_standard) | - |
| Avaya T1 Digital telephones (TH13.21D) | Bootloader: V02_00 Software: V06.02/E27.25/P00.00 |
| Windows-Server, (“Siemens Primergy TX150 S3”) | OS: Windows 2003-Server, Standard Edition Service Pack 1 |
| Avaya CTI-Server software and CTI-administration tools (“ConneCTIon”) | Version V3.1.033 |
| Avaya CTI-TTrace console (software trace tool) | Version 1.20.1.1 |
| Avaya CTI-Modulemanager software (for physical connections) | Version 4.332 |
| Avaya CTI-License server | Version 2.16 |
| Avaya Turret (TH93) | - |
| Integral System Management (ISM) | Version 12.0006 |
| ACB (Advanced Computer Board) | Article Code: 49.9908.9941E |

3. Configure the Avaya I55 PBXs

The configuration of the Avaya Integral 55 PBXs is done via the Integral System Management (ISM) and its components, which are running on a Service PC connected to the systems via LAN. ISM is the basic service tool for administrating the Integral 55 systems. It is an application running under Windows-2000 or Windows-XP operating system. It consists of several components for administrating the different boards of the system.

3.1. Configure of the S0-loop

In order to record calls between VoIP telephones inside one PBX without IP-sniffing, it is necessary to configure a “Basic Rate Interface-loop” (BRI-loop or S0-loop) on the DT0 circuit pack of the Integral 55:

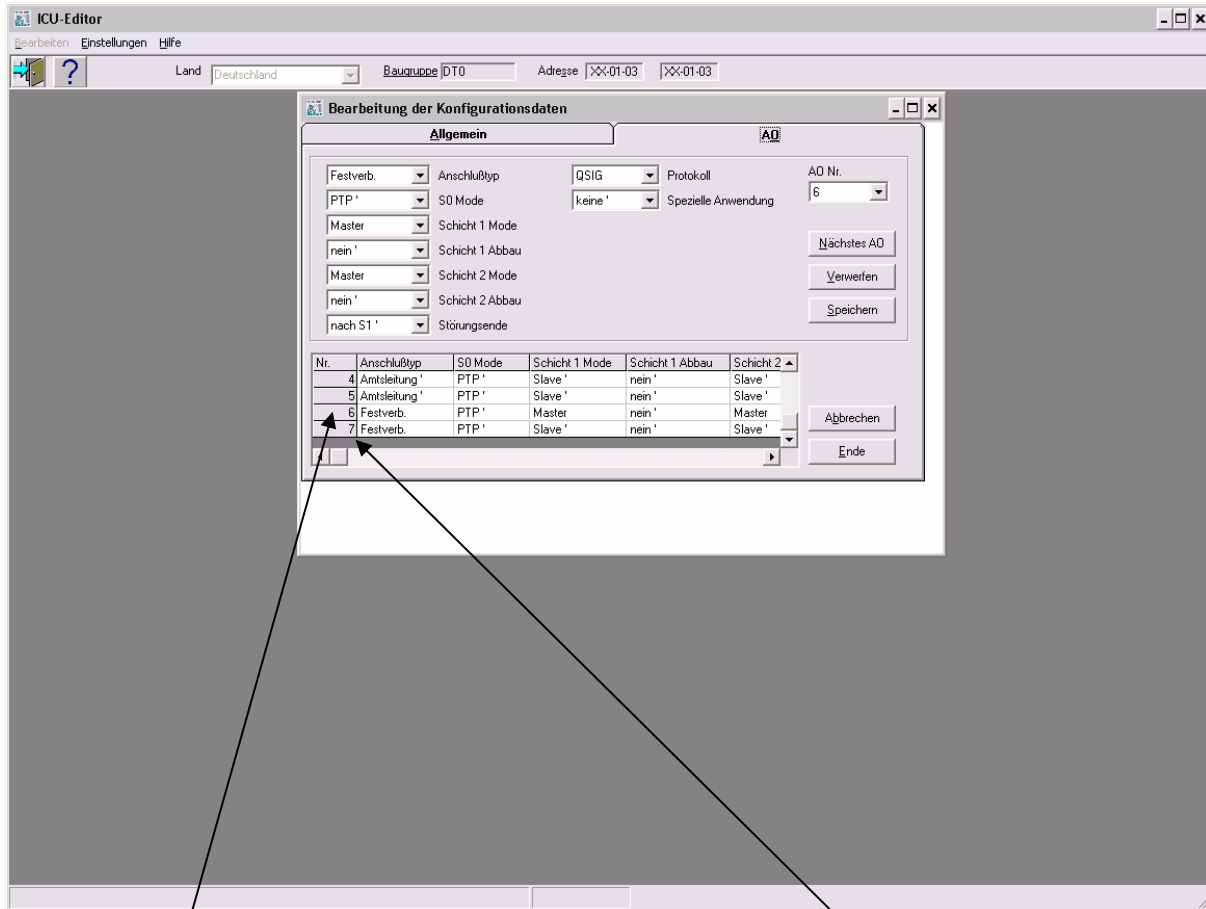
The DT0 board provides 8 digital connections. These can be in the following form:

- 8 T0 interfaces (for public trunks)
- 8 S0 interfaces (for locally-powered terminals)
- 8 S0-FV interfaces for permanent connections (clock master or clock slave) in private connections

For the S0-loop one S0 interface of the DT0 is connected to another S0-interface by using a crossover cable so that the shuffled media streams of pure VoIP calls are routed via the normal switching matrix of the Integral 55. Therefore the MAC-circuit pack, where the ASC Marathon Evolution is connected to, is able to register the voice data and transfer it to a PCM30 timeslot of its E1 output.

The two endpoints of the S0-loop were established at the AO-("Anschlussorgan-") number 6 and 7 on the DT0 board. The hardware address of the DT0 board was "01-01-03".

As protocol for the S0-loop QSIG was chosen. The other settings in the ICU-editor are shown below:



S0-loop endpoint 1:

- Hardware address = 01-01-03-06
- S0 mode = point-to-point connection
- Layer 1 mode = Master

S0-loop endpoint 2:

- Hardware address = 01-01-03-07
- S0 mode = point-to-point connection
- Layer 1 mode = Slave

Via “Transparent Console” (TCO) and MML (Man Machine Language) dialogue the “AOLM” (“Anschlußorgan-Leistungsmerkmale”) program has to be called. Here the following AO-features for the trunks of the S0-loop have to be enabled:

```
AO-Nummer:  AO - Leistungsmerkmale ( Dienst : TLP ):
```

```
-----  
E6000      AMT   DQV   SPR   PRE   CRF   QMS   QIS  
E6100      AMT   DQV   SPR   PRE   CRF   QMS   QIS
```

For the telephones to be monitored, the feature „SPR“ (“Sprachaufzeichnung” meaning call recording) had to be set in AOLM task via MML console additionally.

3.2. Configure the IP-QSIG trunk between the PBXs

For the interconnection between the two Integral 55 systems an IP-QSIG trunk is used. Here the QSIG protocol is tunneled via an IP connection between the VoIP boards of the PBXs. On each side of the IP-QSIG connection an additional trunk had to be established with the following features (via “Transparent Console” (TCO) and MML (Man Machine Language) dialogue):

```
AO-type:           BAN  
Protocol:          QSIG, version = 0  
IP address:        remote IP address of counterpart PBX  
active coder:      g729annexa  
remote calling number: pseudo calling number of the trunk in the counterpart PBX  
services:          TLP, GEN  
B-channel-data allocation: NSTA  
BCH access right:  M (“mit”)  
direction of BCH allocation: W (“wechselseitig”)
```

For the trunks the following AO- (“Anschlussorgan-”) features have to be enabled in the AOLM task:

```
AMT  
DQV  
PRE  
CRF  
QMS
```

3.3. Configure the monitored timeslots of the PCM30 connection

The 30 B-channels (PCM30) of the E1-interface from the MAC board were configured with the “Integral Data Management tool” (IDM).

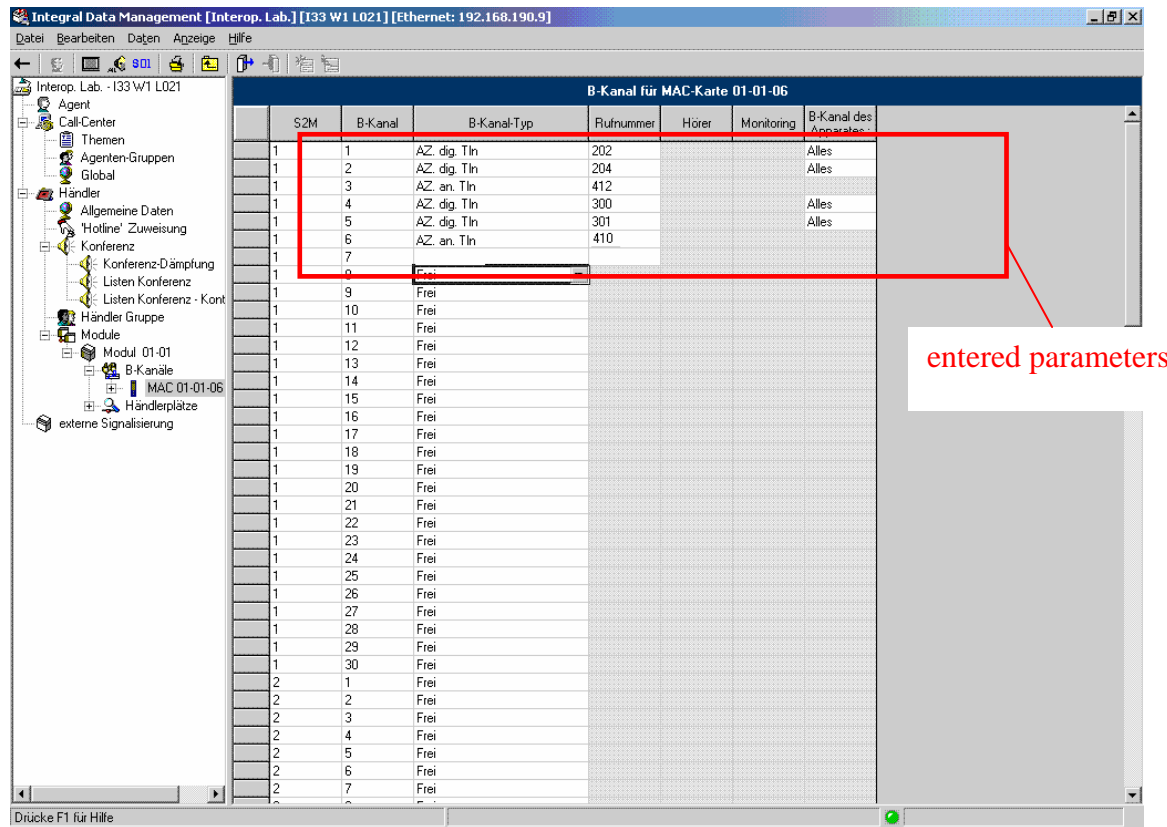
IDM is integrated in the ISM tool suite and can only be used as “superuser”.

The following screenshot shows, how IDM settings were done, to assign dedicated extensions or trunks to a timeslot (B-channel) of the MAC board:

- digital phone, with number “202” is assigned to B-channel 1
- digital phone, with number “204” is assigned to B-channel 2
- digital phone, with number “300” is assigned to B-channel 4
- digital phone, with number “301” is assigned to B-channel 5
- analog phone, with number “412” is assigned to B-channel 3
- analog phone, with number “410” is assigned to B-channel 6

Line one means, that the voice data of the digital phone with the number “202” is delivered in B-channel 1 of the PCM30 signal to the ASC Marathon Evolution server. Thus the 1st recording channel of ASC Marathon Evolution server records all phone calls of extension 202.

For recording the turret TH93, its hardware number had to be entered here too. As B-channel type (“B-Kanal Typ”) the value “AZ Hörer” was chosen (not shown in the screenshot).



4. Configure the ASC Marathon Evolution

As shown in **Figure 1**, ASC Marathon Evolution consists of two major components: the ASC Marathon Evolution server and the ASC CTI Controller server.

The following description of the configuration describes the non-defaulted parameters.

4.1. Configure the ASC CTI Controller server

This configuration is saved in the “ctic.ini” file on the ASC CTI Controller server. It has configuration parameters like the IP address of the used license server, and the used alarm manager. Additionally, login parameters for the Avaya CTI server and the extensions to be captured are configured here.

```
[GENERAL]
;AgentRecoveryMode=Off|NoTimeout|WithTimeout
AgentRecoveryMode=Off
;Checks every [AgentLoginCheckInterval] minutes, if agents should timeout
after [AgentLoginTimeout] minutes
AgentLoginCheckInterval=60
; Interval in minutes, after which agents get logged out automatically
AgentLoginTimeout=10080

[LogModule]
Global Level      = 3
Logfile Level    = 3
Logfile Size     = 4000000
Console Level    = 3
Eventlog Level   = 3
Debugout Level   = 3

[AlarmMan]
Address = localhost
Port = 0
Send = 60000
Wait = 10000
ReconnectIntervall = 1000

[Licenseserver]
Address = 192.168.1.30
Port = 7000
Protocol = CCOPEN
Send = 60000
Wait = 10000
ReconnectIntervall = 1000

[CONTROLLER-1]
ControllerName=TenovisController
ControllerDll=CTIC_CTRL_Tenovis.dll
EssServerAddress=192.168.190.110
EssServerPortNo=50050
EssClientAddress=CTICLORENZ
```

← address data of the used
license server

```
EssClientPortNo=10000
EssClientApiDllName=ESSClientAPI.dll
StringMultiplePartners=MULTIPLE
StringNotAvailable=N/A
```

```
[PBX-1]
PbxID=1
PbxName=Tenovis PBX
DriverIDForPbx=1
ServiceName=Tenovis CTI-Server
Login=admin
Password=admin
```

access data for the CTI server



```
[PBXDRIVER-1]
DriverName=TenovisTsapiDriver
DriverDll=CTIC_PBXD_TenovisTSAPI.dll
RecoveryTimerInterval=40000
```

// This is how a monitor point is set on a trunk

```
[MONITORPOINT-1]
ControllerID1=1
PbxID=1
MonitorDeviceID=300
RecordingMode=ES
DeviceType=Extension
```

data for the extensions to be recorded



// This is how a monitor point is set on a trunk

```
[MONITORPOINT-2]
ControllerID1=1
PbxID=1
MonitorDeviceID=301
RecordingMode=ES
DeviceType=Extension
```



```
[MONITORPOINT-3]
ControllerID1=1
PbxID=1
MonitorDeviceID=202
RecordingMode=ES
DeviceType=Extension
```



```
[MONITORPOINT-4]
ControllerID1=1
PbxID=1
MonitorDeviceID=204
RecordingMode=ES
DeviceType=Extension
```



```
[MONITORPOINT-5]
ControllerID1=1
PbxID=1
MonitorDeviceID=412
RecordingMode=ES
DeviceType=Extension
```

```
[MONITORPOINT-6]
ControllerID1=1
PbxID=1
MonitorDeviceID=410
RecordingMode=ES
DeviceType=Extension

//This is the monitor configuration of the 1st B-Channel of the turret
10000.
[MONITORPOINT-14]
ControllerID1=1
PbxID=1
MonitorDeviceID=4003#1
RecordingMode=ES
DeviceType=Extension

//This is the 2nd B-Channel of turret 10000 (Handset 2)
[MONITORPOINT-15]
ControllerID1=1
PbxID=1
MonitorDeviceID=4003#2
RecordingMode=ES
DeviceType=Extension

[MONITORPOINT-16]
ControllerID1=1
PbxID=1
MonitorDeviceID=D1000
RecordingMode=ES
DeviceType=Trunk

[MONITORPOINT-17]
ControllerID1=1
PbxID=1
MonitorDeviceID=D1001
RecordingMode=ES
DeviceType=Trunk

[ESCHNL-1]
ControllerID=1
MarathonID=0
ChannelID=4
ExtensionMap=300#1
ControlMode=SS
PbxID=1

[ESCHNL-2]
ControllerID=1
MarathonID=0
ChannelID=5
ExtensionMap=301#1
ControlMode=SS
PbxID=1

[ESCHNL-3]
ControllerID=1
```

MarathonID=0
ChannelID=1
ExtensionMap=202#1
ControlMode=SS
PbxID=1

[ESCHNL-4]
ControllerID=1
MarathonID=0
ChannelID=2
ExtensionMap=204#1
ControlMode=SS
PbxID=1

[ESCHNL-5]
ControllerID=1
MarathonID=0
ChannelID=3
ExtensionMap=412#1
ControlMode=SS
PbxID=1

[ESCHNL-6]
ControllerID=1
MarathonID=0
ChannelID=7
ExtensionMap=410#1
ControlMode=SS
PbxID=1

[ESCHNL-7]
ControllerID=1
MarathonID=0
ChannelID=8
ExtensionMap=4003#1
ControlMode=SS
PbxID=1

[ESCHNL-8]
ControllerID=1
MarathonID=0
ChannelID=9
ExtensionMap=4003#2
ControlMode=SS
PbxID=1

4.2. Configure the ASC Marathon Evolution server

The configuration for recording the hardware channel 1 is shown in **Figure 2**.

Set the “RecordStartMode” to “HOST”.

This means the external CTI-Controller application is responsible for supplying event information indicating that a voice call should be recorded.

Set the “Compression” field to “PCM_A_LAW”.

The input parameters “InputSource1” and “InputType1” must be configured correctly.

Set the “InputSource1” field to “COMMAN (Analog / PCM30)”, which identifies the physical E1 card to be used for recording.

Set the “InputType1” field to “PRI_ACTIVE_TIMESLOT” for the” which means the recorder is terminating the E1 trunk in an active fashion.

Set the “InputSlot1” field to the timeslot for the channel.

ASC

ASC DataManager 1.7

- ASC DataManager
- User Administration
- Configuration
 - System
 - Channels**
 - EVOip channels
 - Auto Tagging
 - Channel guard
 - Recording Planne
 - Recorder informa
 - Recorder informa
- Archive Client
- SDDM Client
- Registry
- Information

Channels

| State | ChannelDescription | ChannelID |
|-------|--------------------|-----------|
| OK | Channel 001 | 0AG9MU001 |
| OK | Channel 002 | 0AG9MU002 |
| OK | Channel 003 | 0AG9MU003 |
| OK | Channel 004 | 0AG9MU004 |
| OK | Channel 005 | 0AG9MU005 |
| OK | Channel 006 | 0AG9MU006 |

Configuration of Channel 001

| State | Name | Description | Value(s) (De-/Select all) | |
|-------------------------------------|-------------------------|---|---|-------------------------------------|
| | RecordStartMode | Start recording by: | HOST (External application) CONTINUOUS (Always recording.) VOX (Signal level) COR (Contact operation) | <input checked="" type="checkbox"/> |
| | RecordStopMode | Stop recording by: | - (Use the triggers from recording start) HOST (External application) VOX (Signal level) COR (Contact operation) | <input checked="" type="checkbox"/> |
| <input checked="" type="checkbox"/> | StorageMode | Storage mode | COMPLETE_CALL_INFO (Store when all call | <input checked="" type="checkbox"/> |
| | VoxLevel | Threshold value for sensitivity of signal detection. Range from 0dB (max sensitive) to 62dB (least sensitive). | 20 dB | <input checked="" type="checkbox"/> |
| | Timespan_Until_Deletion | Time to keep a call in the database (YY:MM:DD:HH:mm). | 99:00:00:00:00 | <input checked="" type="checkbox"/> |
| | CLIEnable | Enable CLI detection | No | <input checked="" type="checkbox"/> |
| | DTMFEnable | Enable DTMF detection | No | <input checked="" type="checkbox"/> |
| | PreTrigger | PreTrigger to use by record start [0..51]*100ms. | 20 | <input checked="" type="checkbox"/> |
| | Compression | Compression to use for audio data | PCM_A_LAW (PCM_A_LAW) | <input checked="" type="checkbox"/> |
| | VoxPostTime | Minimum duration for silence before recording stop in conjunction with VOX trigger. 100ms+[0..1023]*100ms | 79 | <input checked="" type="checkbox"/> |
| | VoxTimeMin | Minimum signal duration before recording start in conjunction with VOX trigger. | 1000 ms | <input checked="" type="checkbox"/> |
| | IdlePostTime | Minimum duration for silence before recording stop in conjunction with IDLE WORD trigger. 100ms+[0..1023]*100ms | 49 | <input checked="" type="checkbox"/> |
| | IdleTimeMin | Minimum signal duration before recording start in conjunction with IDLE WORD trigger. | 500 ms | <input checked="" type="checkbox"/> |
| | PackageTimeout | Time to wait before call packages get finally processed after call end. Unit is 100 ms. | 100 | <input checked="" type="checkbox"/> |
| | AGCEnable | Enable AGC mode. | Enabled (Mono) | <input checked="" type="checkbox"/> |
| | ActiveHook | Take and record analog PBX-conference calls | Off | <input checked="" type="checkbox"/> |
| | BeepToneEnable | Beep tone insertion. | Off | <input checked="" type="checkbox"/> |
| | AnalogGain | Gain for analog lines. | 0 dB | <input checked="" type="checkbox"/> |
| | AGCRaiseTime1 | AGC raise time for the first input channel. | 608 ms | <input checked="" type="checkbox"/> |
| | AGCMaxGain1 | AGC maximum gain for the first input channel. | 41 dB | <input checked="" type="checkbox"/> |
| | InputSource1 | Type of recording interface | COMMAN (Analog / PCM30) | <input checked="" type="checkbox"/> |
| | InputType1 | Signal Input | PRI_ACTIVE_TIMESLOT (Active PRI input) | <input checked="" type="checkbox"/> |
| | InputSlot1 | The time slot number of the recording interface | 1 | <input checked="" type="checkbox"/> |
| | InputSource2 | Type of correspondent recording interface | COMMAN (Analog / PCM30) | <input checked="" type="checkbox"/> |
| | InputType2 | The InputType of the second InputSource. | DISABLED (Disabled) | <input checked="" type="checkbox"/> |
| | InputSlot2 | The time slot number of the correspondent recording interface | 2 | <input checked="" type="checkbox"/> |
| | Availability | This channel is physically available | Yes | <input checked="" type="checkbox"/> |

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Figure 2: Configuring PCM30 channel

For the IP-sniffing method to monitor an IP extension, configure the “EVOip channels” in the ASC Data Manager tool as shown in **Figure 3**.

Set the “RecordStartMode” to “HOST”. This means the external CTI-Controller application is responsible for supplying event information indicating that a voice call should be recorded. Set the “RecordStopMode” to “Use the triggers from recording start”, to use the same trigger for stopping recording a call as for starting.

Set the “StorageMode” field to “COMPLETE_CALL_INFO”, which records additional information from CTI at the end of the captured voice data.

Leave the “Timespan_Until_Deletion” field at the default value of “99:00:00:00:00”, meaning that the recorded data will be deleted in 99 years

Leave the “InputType1” field at the default value of “STATIC_STEREO_STREAMS”, meaning that a data stream is expected as input.

Enter the IP address of the phone, which should be captured in the “IP” field.

ASC DataManager 1.7

EVOip channels

| State | ChannelDescription | ChannelID |
|-------|----------------------|------------|
| OK | DYNAMIC VOIP CHANNEL | N/A |
| OK | EVOip Channel 001 | 4QA7H1M3UX |
| OK | EVOip Channel 002 | 4QA7H1M3UY |
| OK | EVOip Channel 003 | 4QA7H1M3UZ |
| OK | EVOip Channel 004 | 4QA7H1M3V0 |
| OK | EVOip Channel 005 | 4QA7H1M3V1 |
| OK | EVOip Channel 006 | 4QA7H1M3V2 |
| OK | EVOip Channel 007 | 4QA7H1M3V3 |
| OK | EVOip Channel 008 | 4QA7H1M3V4 |
| OK | EVOip Channel 009 | 4QA7H1M3V5 |

Configuration of EVOip Channel 001

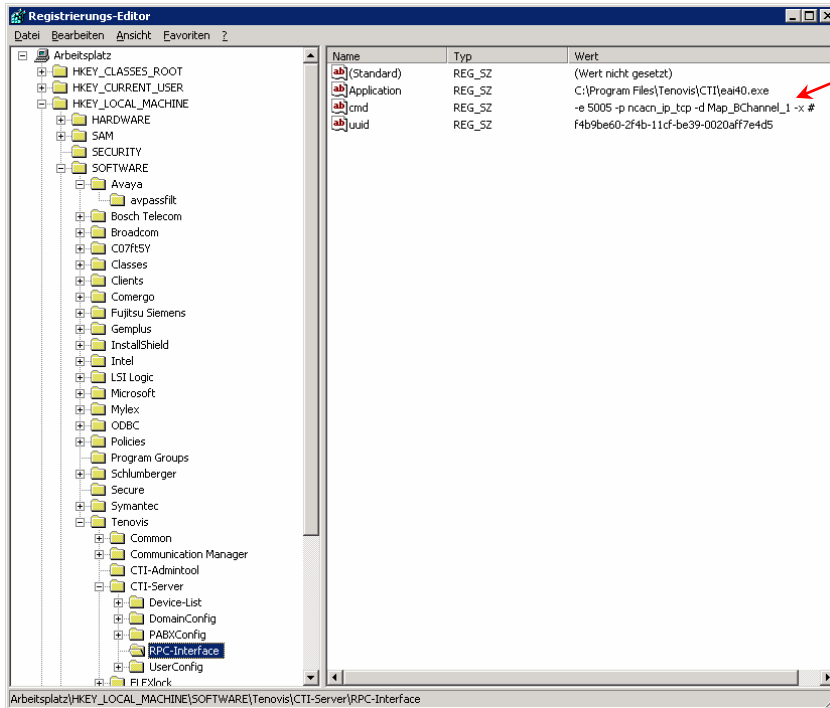
| State | Name | Description | Value(s) (De-/Select all) |
|-------|-------------------------|---|---|
| | RecordStartMode | Start recording by: | HOST (External application) ALERTING (Conversation indication) CONNECT (Conversation detection) DTMF-SEQUENCE (DTMF sequence) |
| | RecordStopMode | Stop recording by: | - (Use the triggers from recording start) HOST (External application) CONNECT (Conversation detection) DTMF-SEQUENCE (DTMF sequence) |
| | StorageMode | Storage mode | COMPLETE_CALL_INFO (Store when all call |
| | Timespan_Until_Deletion | Time to keep a call in the database (YY:MM:DD:HH:mm). | 99:00:00:00:00 |
| | PackageTimeout | Time to wait before call packages get finally processed after call end. Unit is 100 ms. | 100 |
| | InputType1 | The input type of this channel. | STATIC_STEREO_STREAMS (STATIC_STER |
| | IP | The IP address of the input. | 192.168.190.51 |
| | MacAddress | The Mac address of the input. | |
| | Extension | The Extension of the input. | |
| | Availability | This channel is physically available | Yes |

Figure 3: Configuring an IP channel for IP-sniffing

4.3. Configure the Avaya CTI Server

The Avaya CTI server has been installed according to the user manual provided by Avaya. The following setting differed from the default parameters:

- The device ID mapping in the CTI server options was mandatory. Therefore a parameter had to be added to the “cmd” entry in the Windows registry of the server:
“HKEY_LOCAL_MACHINE\SOFTWARE\Tenovis\CTI-server\RPC-Interface: -d Map_BChannel_1 -x #”



5. Interoperability Compliance Testing

5.1. General Test Approach

Testing included validation of correct operation of typical voice recording functions including:

- Calls inside one PBX (basic call, supplementary services, threat call)
- Calls between two networked PBXs (basic call, supplementary services)
- Calls from external PSTN (basic call, supplementary services)
- Error and recovery treatment

5.2. Test Results

All tests passed.

6. Verification Steps

To verify that the solution was properly configured, the following steps can be taken:

After establishing the physical connection to the CTI server via LAN, verify that the CSTA Make-Call command works properly via the Avaya “CTI Administrator” tool.

To verify the correct physical connectivity between the MAC circuit pack and the ASC Marathon evolution server, a LED on the backside of the server must be lit.

7. Support

For technical support for the ASC Marathon Evolution solution, please contact the technical support hotline of ASC:

Email: hotline@asc.de

8. Conclusion

These Application Notes describe the configuration steps required for ASC Marathon Evolution to successfully interoperate with Avaya Integral 55. A Linux based Advanced Computer Board (ACB) running software version L021 was used. Recording of telephone calls in a single PBX and between networked PBXs were validated either and the error and recovery treatment of the solution was checked. The configuration described in these Application Notes has been successfully compliance tested.

9. Additional References

[1] Additional product information from ASC
http://www.asctelecom.com/english/index_e.html

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